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GB A 2137458	GB 1511487
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GB A 2105153	GB 1444828
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GB 1550721	GB 1232974
GB 1542700	GB 1058556
GB 1524756	WO A1 8403407
GB 1519607	

(58) Field of search
H4R
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(54) Loudspeaking telephone

(57) In a loudspeaking telephone the transmit channel and the receive channel include respective attenuators (AT3, AT7) which are controlled by a microprocessor (M11) in accordance with the relative AC signal strength in those channels. This is done by tapping off these signals via analogue-digital converters (C10, C12) whose outputs go to the microprocessor (M11). This assesses which is the active channel and adjust the attenuators (AT2, AT7) accordingly.

In the idle or standby state, both attenuators are set at their mid-points, and when the attenuator settings have to be altered, the attenuation is decreased step-wise in the active channel and increased step-wise in the passive channel. The increases and decreases are so done that the sum of the attenuations provided by the two attenuators is substantially constant and the switching threshold is adjusted according to the coupling between the channels.

Fig.1.

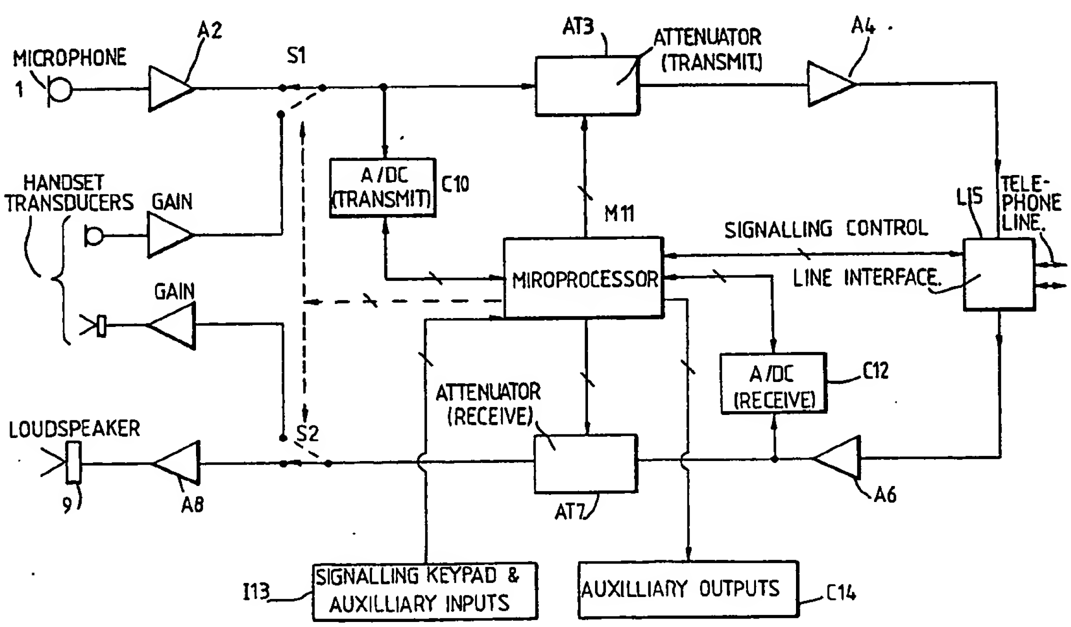

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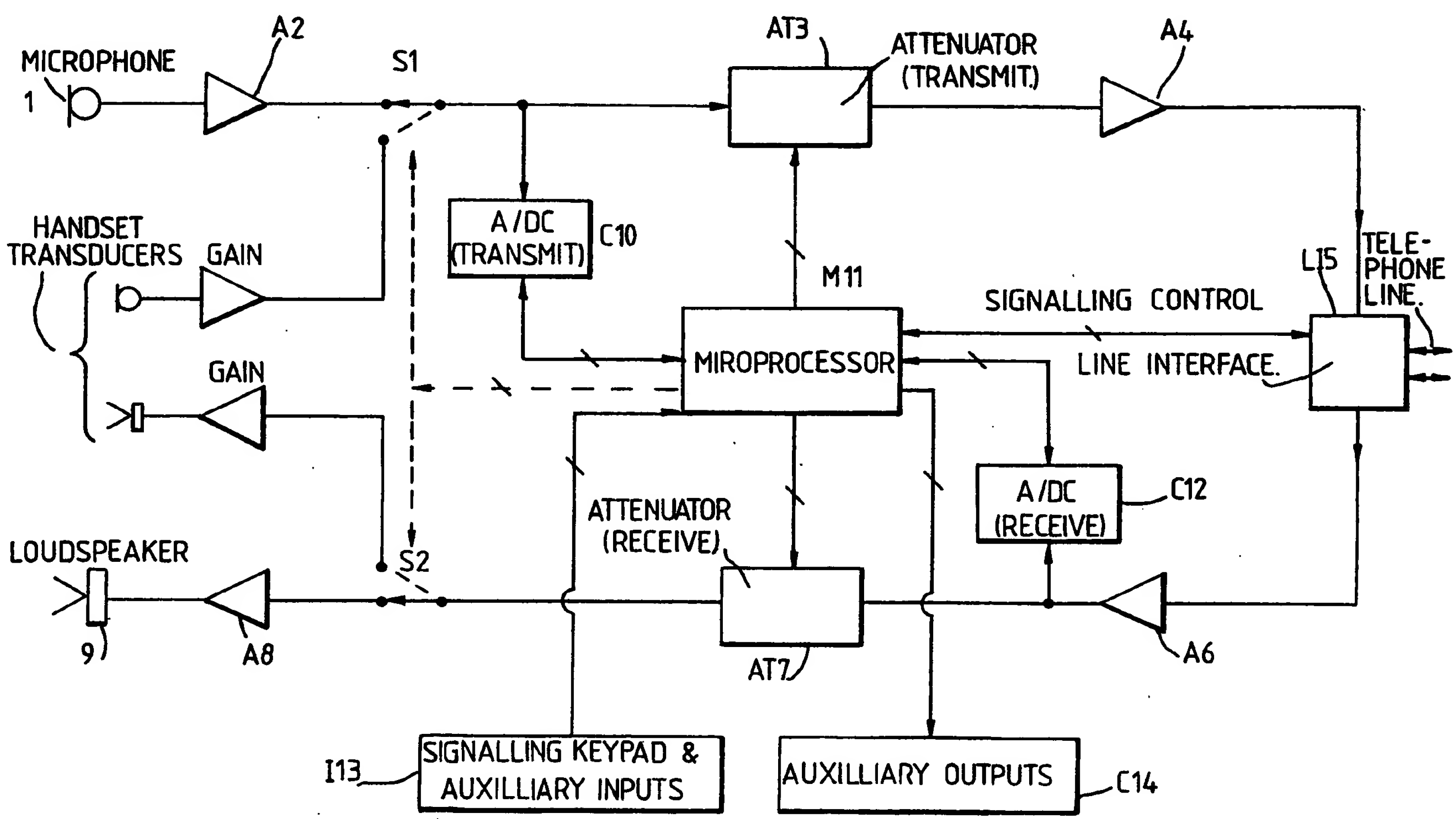
Fig.1.



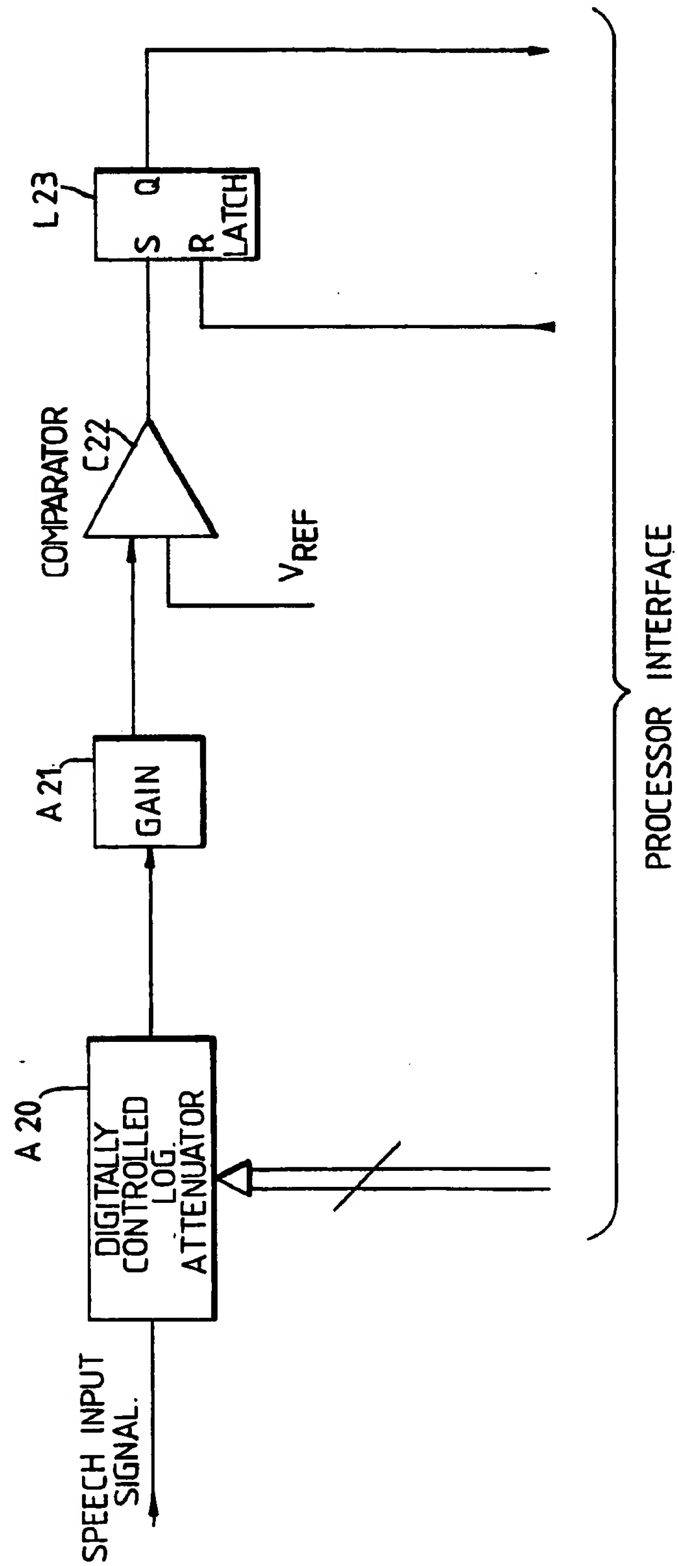
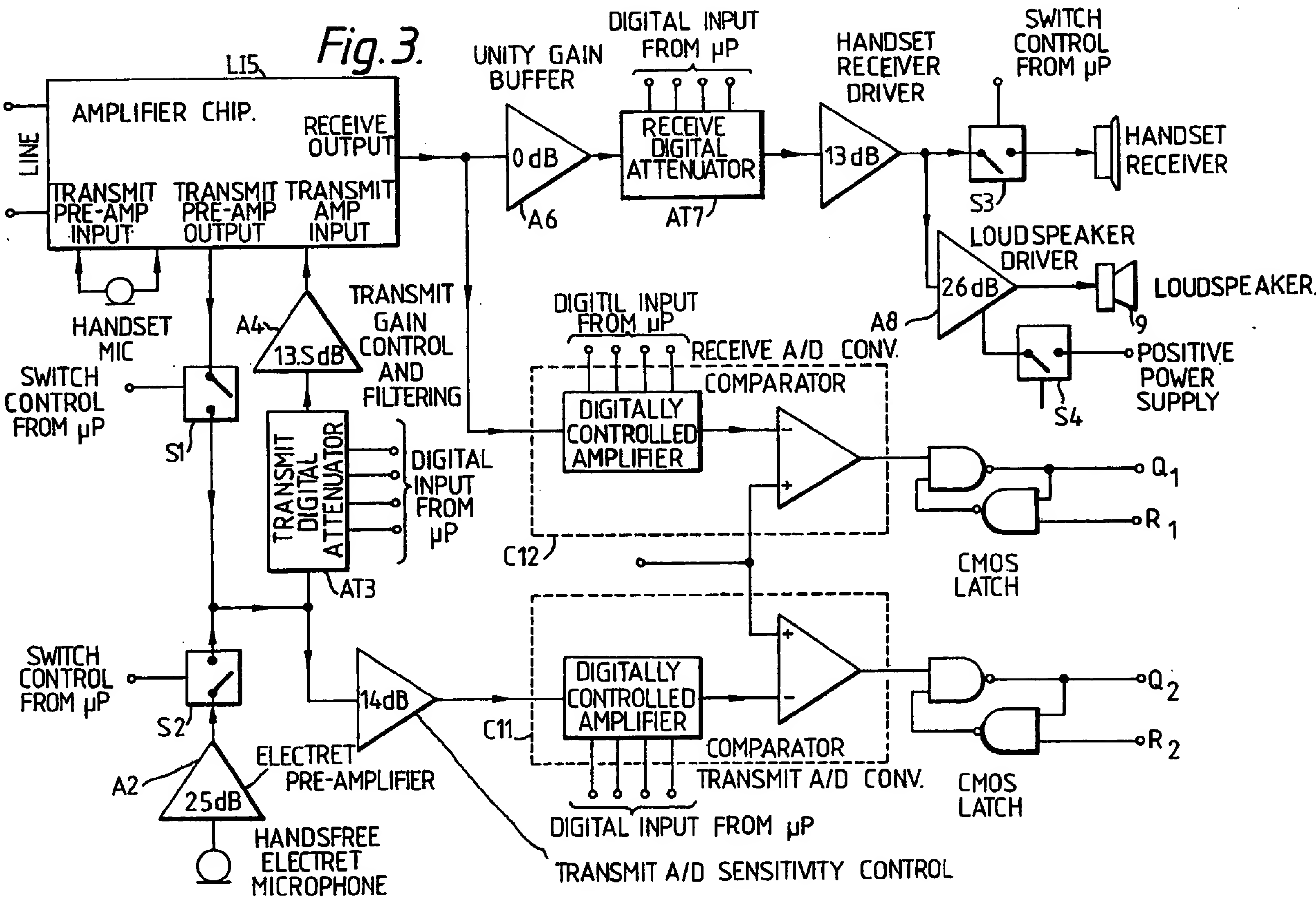

SPEECH INPUT SIGNAL.
A 20
DIGITALLY CONTROLLED LOG. ATTENUATOR
A 21
GAIN
COMPARATOR
C22
V_{REF}
L 23
S
Q
R
LATCH
PROCESSOR INTERFACE

Fig.2.

Fig. 3.


L15
AMPLIFIER CHIP.
LINE
RECEIVE OUTPUT
TRANSMIT PRE-AMP INPUT
TRANSMIT PRE-AMP OUTPUT
TRANSMIT AMP INPUT
HANDSET MIC
SWITCH CONTROL FROM µP
S1
A4
13.5dB
TRANSMIT GAIN CONTROL AND FILTERING
TRANSMIT DIGITAL ATTENUATOR
AT3
DIGITAL INPUT FROM µP
SWITCH CONTROL FROM µP
S2
A2
25dB
ELECTRET PRE-AMPLIFIER
HANDSFREE ELECTRET MICROPHONE
14dB
TRANSMIT A/D SENSITIVITY CONTROL
UNITY GAIN BUFFER
0 dB
A6
DIGITAL INPUT FROM µP
RECEIVE DIGITAL ATTENUATOR
AT7
HANDSET RECEIVER DRIVER
13 dB
SWITCH CONTROL FROM µP
S3
HANDSET RECEIVER
LOUDSPEAKER DRIVER
26 dB
A8
LOUDSPEAKER.
9
POSITIVE POWER SUPPLY
S4
DIGITIL INPUT FROM µP
RECEIVE A/D CONV.
COMPARATOR
DIGITALLY CONTROLLED AMPLIFIER
C12
Q1
R1
CMOS LATCH
C11
DIGITALLY CONTROLLED AMPLIFIER
COMPARATOR
TRANSMIT A/D CONV.
DIGITAL INPUT FROM µP
Q2
R2
CMOS LATCH

SPECIFICATION

Loudspeaking telephone

This invention relates to electronic circuits for use in loudspeaking or handsfree telephones.

In our Appln. No. 8216845 (Serial No. 2122851A) (P. F. Blomley 8) we have described a handsfree telephone system to which the present invention may be applied. In that system, the handsfree microphone is connected via a transmit channel including an amplifier, an attenuator and another amplifier to the line interface. The line interface is connected via a receive channel including an amplifier, another attenuator and yet another amplifier to the loudspeaker. The signals in the two channels are passed via respective analogue-digital converters to a comparator which assesses from the relative signal amplitudes which is the active and which the passive channel. It then decreases the attenuation in the active channel. This avoids the "singing" due to coupling between the two channels which is the bane of many loudspeaking or handsfree systems.

An object of the invention is to provide improvements to the system of the above-mentioned application, although it will be appreciated that the present invention is applicable to other loudspeaking or handsfree sets.

According to the invention, there is provided an electronic circuit for use in a loudspeaking or handsfree voice telephone system which includes a first analogue-digital converter to which the signal in the system's transmit channel is applied and which derives therefrom a succession of digital combinations each representing the current amplitude of the AC signal in that channel, a second analogue-digital converter to which the signal in the systems receive channel is applied and which derives therefrom a succession of digital combinations each representing the current amplitude of the AC signal in the receive channel, comparator means to which the digital combinations from the two converters are applied and which derives therefrom a control output which indicates which of the two channels is to be activated, a first attenuator in the transmit channel, a second attenuator in the receive channel, and control connections from the comparator means by which said control output controls the attenuator via said control connections, the control being such that

(a) in the standby condition in which there is little or no AC signal in either channel, the attenuators are set to a condition midway or approximately midway between their highest and their lowest values,

(b) if a speech signal is detected in one channel and not in the other, or if a speech signal in one channel has an amplitude which is larger by at least preset threshold than the speech signal in the other channel, the attenuators are so adjusted that the value of the attenuation in said one channel is decreased in a step-wise manner while the value of the attenuation in the other channel is increased in a step-wise manner,

(c) the above threshold is altered in accordance with the level of coupling between the two channels, and

(d) the overall sum of the attenuations in said two channels remains substantially constant.

An embodiment of the invention will now be described with reference to the accompanying drawings, in which

Figure 1 is a highly simplified block diagram of a handsfree telephone embodying the invention.

Figure 2 is a simple schematic of part of the circuitry of Figure 1.

Figure 3 is a more detailed block diagram of the arrangements shown in Figure 1, but excluding the microprocessor.

From these drawings, the similarities between the embodiment described and the system of our above-mentioned application will be readily apparent.

We now consider the simple block diagram, Figure 1. Here we see the handsfree microphone, which in this case is an electret microphone connected via na amplifier A2, a switch S1 shown in the position for handsfree operation, an attenuator AT3 and another amplifier A4 to a line interface L15. This gives the set access to a telephone line.

The line interface L15 is also connected via an amplifier A6, attenuator AT7, switch S2 shown set for handsfree operation, and another amplifier A8 to the loudspeaker 9.

The handset transducers are also connected via amplifiers to contacts of S1 and S2, which are contacts of a switch in the telephone which is manually settable to the "handset" or "handsfree" condition.

The input to AT3 is also connected via an analogue-digital converter C10 to a microprocessor M11, which is also connected to S1 and S2. Hence changes in the settings of S1 and S2 are controlled by M11. This microprocessor can also control the attenuators AT3 and AT7 via the dashed-line connections shown. The microprocessor also has an input from another analogue-digital converter C12 via which the incoming speech signals from the remote exchange are noted. It also has inputs from the instrument's keyboard and other inputs I13, and auxilliary outputs C14.

The attenuators have a range of 0 to 60 dB, in 4 dB steps, adjustments to their settings being made under control of the microprocessor M11. In the rest or standby state the attenuators are both set to the midpoints of their ranges, i.e. to 30 dB. During operation the AC (speech) levels in the transmit and receive paths are each connected by the converters C11 and C12 into digital words at preset intervals and each pair of words is compared by the microprocessor M11, which thus determines which is the active channel. Then the microprocessor causes the attenuation to be decreased in the active channel and increased in the passive channel. The reduction in attenuation in the active channel and the increase in the passive channel are so

done that the sum of the two attenuations value is constant. Thus the attenuation of the loop which includes the microphone and the loudspeaker is substantially constant.

In addition to the above, the attenuation for the receive channel is modified in accordance with the telephone set's volume control setting, and that of the both channels are modified in accordance with the noise guard.

Thus we have three states for the system:

(a) *Standby* Neither channel is active, and the attenuators are at their nominal central settings.

(b) *Active* Speech is present and the active path is in the nominal *on* state, with the passive path attenuation determined by the sum of the two attenuations.

When speech ceases on the active channel the attenuators remain unchanged for a time TH, approximately 200 ms, to avoid speech clipping, before Rampdown.

(c) *Rampdown* Active and passive path attenuations are increased and reduced, respectively, until the *Standby* state is reached, or a channel becomes active. The rate of change is 40 ms per 4 dB.

Note that when speech is detected, the appropriate channel is switched *on* immediately, to avoid clipping.

Figure 2 shows in block form one of the analogue-digital converters, C10 and C11 of Figure 1. Here the speech input is applied to a digitally controlled logarithmic attenuator A20 which is controlled from the microprocessor. The output of A20 is applied via an amplifier A21 to one input of a comparator C22 where the analogue signal value is compared with successive reference voltages in normal manner. The output of this comparator goes to a latch L23 whose output is thus a sequence of digits (1 or 0) which correspond to the input speech amplitude. This latch is controlled from and supplies its output to the microprocessor, where the comparison between the signals in the two channels is effected.

We now refer to Figure 3, which shows in more detail the arrangement of Figure 1. Where appropriate the references used in Figure 1 have been used again in Figure 2. The dB values given for the various amplifiers are those used in one specific example of telephone circuit. There are also shown four switches S1, S2, S3 and S4, all controlled from the microprocessor. Both attenuators are digitally controlled from the microprocessor, and the digitally controlled amplifiers in the convertor blocks correspond to the A20-A21 combination in Figure 2.

The two latches are shown as latches whose outputs provide the two sets of digital amplitude samples which are used by the processor in controlling the circuit.

Much of the operation of the circuit is controlled by the microprocessor's software, and this operation will now be considered.

With handsfree telephones there is a risk of "wrong path switching", e.g. a received speech signal may be acoustically coupled to the microphone. It may then be wrongly accepted as a genuine speech signal from the microphone, which could cause erroneous switching, thus cutting off a remote talker.

To overcome this difficulty a preset margin is allowed by which the transmit signal has to exceed the coupled received signal before reversal of directions occurs. Thus a differential threshold DFT, defined by the sum of the receive path-transmit path coupling and this margin, is specified. This threshold DFT is varied by the microprocessor in accordance with the setting of the receive path attenuator.

Similarly there is a differential threshold DFR which takes account of sidetone coupling.

These various constants and thresholds are all taken into account by the microprocessor's software as it controls the switching operations effected when the handfree telephone set is in use.

Note that the differential test to be done under microprocessor control is only applied to the currently off path.

In operation, the two thresholds referred to above, which determine the switching between the transmit and the receive channels, respectively track the attenuators' settings to give a constant wrong path switching margin under all conditions.

The *standby* state calls for two more thresholds, upper UT and lower LT, where

$$LT = UT - DT$$

DT is constant hysteresis term.

These thresholds may have different values for transmit and receive. Hence for a path to become active, either from standby or due to reversing direction, the speech envelope value must equal or exceed the value of UT as well as that of DFT or DFR as appropriate.

To avoid spurious switching due to room noise, the value of UT for transmit is increased by the measured transmit noise level TNL such that the required threshold is above noise level. Note also that room noise necessitates increased received volume, as a function of TNL. In addition, the normal transmit on channel attenuation is increased with increasing room noise to compensate for the fact that people tend to speak louder in the presence of background noise.

The value of TNL used, as indicated above, is derived from the envelope measurement of the transmit path signal. This is effected in response to the various digital sample values determined by the processor.

It has been found that the spectrum of the speech envelope peaks at about 5Hz, probably the syllabic rate, and is about 16 dB less at 60 Hz. For use on these TNL determinations, an approximation to a low pass filter has been taken in software by taking the means of the transmit envelope over 16 samples. This, with a main

programme loop time of 1 ms gives a filter frequency of 62 Hz. The higher envelope frequencies are taken out, whereafter the minimum mean envelope MME measured over 1 sec is taken as the background noise over that period. If MME over 1 sec is less than TNL, then TNL is decremented by unity, while if it is greater than TNL, TNL is incremented by unity. Thus TNL is adjustable by 4 dB steps, one per second.

The attenuators are adjusted as a function of TNL, and full adjustment is required to occur at or above a sound pressure level of 57 dBA.

Noise measurement is only effected when the receive channel is *not* active, so that the received signal is not interpreted as noise via the feedback path between the two channels.

The transmit path noise guard cannot be used in the receive path as it would be improperly effected by call progress tones which last longer then 1 sec. Hence the receive channel's UT is varied in accordance with receive signal level, so that on long lines it is minimum, becoming larger on short lines.

Volume control uses two push buttons, one to increase volume and one to decrease it, in each case in steps of 4 dB per button depression.

At this point it is convenient to summarize the speech envelope derivation. For each channel the hardware consists of a digitally controlled (via the processor) attenuator A20-A21, Figure 2, a comparator C22, and reference voltage, and a set/reset latch L23, which is set by the comparator and reset under processor control.

Envelope derivation occurs for the transmit and receive signals at a rate determined by the main programme loop, approximately 1 ms, and is software controlled. The latch serves to record the event of an input signal exceeding the appropriate current envelope level.

At entry into the envelope part of the software, if it is found by testing the latch output that the input signal has exceeded the last digitised value, then this value output to the attenuator stage is incremented one step (4 dB) at a time until the latch output becomes zero. During this process the latch is held reset such that it follows inversely the comparator output. The envelope timer is now set, and the latch taken out of the reset state. The setting of A20 is taken as the current digital value of the speech envelope.

Conversely, if the input does not exceed the envelope value and the time out has expired the envelope is decremented by one step and the timer set.

If a receive speech signal ends abruptly, the wrong path switching margin may be reduced because of the finite propagation time of the directly-coupled acoustic path from loudspeaker to microphone. Thus if the receive signal ends abruptly, the coupled signal referred to continues unchanged for a brief period given by the propagation time mentioned above. It then falls to the room reverberation level and ultimately decays to the ambient noise level. Hence there is an envelope hold time included in the information on which the microprocessor works, so that in the case of such a sudden cut off a reasonable level wrong path switching margin is retained.

The keypad of the instrument has a 16 button keypad, 12 of which are the usual signalling buttons of a push-button set. The others are Volume Up, Volume Down, Handsfree and Mute.

The system has five modes of operation:

(1) Idle - no action, apart from the subscriber keying in a wanted number.
(2) Handsfree - voice switching is applied using the loudspeaker and subset microphone.
(3) Handset - voice switching applied using the handset transducers only.
(4) Monitor - listen only using the loudspeaker, there being no voice switching.
(5) Group Listening - voice switching using the handset transducers and the loudspeaker.

The change from mode to mode depends on the state of the hook-switch, and the operation of the Handsfree and Mute button. These are set by the user in accordance with manufacturer's instructions, and the settings of these switches is monitored by the microprocessor and the results of the monitoring added on to control the circuit.

CLAIMS

1. An electronic circuit for use in a loudspeaking or handsfree voice telephone system, which includes a first analogue-digital converter to which the signal in the system's transmit channel is applied and which derives therefrom a succession of digital combinations each representing the current amplitude of the AC signal in that channel, a second analogue-digital converter to which the signal in the system 's receive channel is applied and which derives therefrom a succession of digital combinations each representing the current amplitude of the AC signal in the receive channel, comparator means to which the digital combinations from the two converters are applied and which derives therefrom a control output which indicates which of the two channels is to be activated, a first attenuator in the transmit channel, a second attenuator in the receive channel, and control connections from the comparator means by which said control output controls the attenuator via said control connections, the control being such that

(a) in the standby condition in which there is little or no AC signal in either channel, the attenuators are set to a condition midway or approximately midway between their highest and their lowest values,

(b) if a speech signal is detected in one channel and not in the other, or if a speech signal in one channel has an amplitude which is larger by at least a preset threshold than the speech signal in the other channel, the attenuators are so adjusted that the value of the attenuation in said one channel is decreased in a step-wise manner while the value of the attenuation in the other channel is increased in a step-wise manner,

(c) the above threshold is altered in accordance with the level of coupling between the two channels, and

(d) the overall sum of the attenuations in said two channels remains substantially constant.

2. An electronic circuit for use in a loudspeaking or handsfree telephone, which includes a first analogue-digital converter to which the signal in the telephone's transmit channel is applied and which derives therefrom a succession of digital combinations each representing the current amplitude of the AC signal in that channel, a second analogue-digital converter to which the signal in the receive channel is applied and which derives therefrom a succession of digital combinations each representing the current amplitude of the AC signal in the receive channel, comparator means to which the digital combinations from the two converters are applied and which derives therefrom a control output which indicates which of the two channels is to be activated, a first attenuator in the transmit channel, a second attenuator in the receive channel, and control connections from the comparator means by which said control output controls the attenuator via said control connections, the control being such that

(a) in the standby condition in which there is little or no AC signal in either channel, the attenuators are set to a condition midway or approximately midway between their highest and their lowest values,

(b) if a speech signal is detected in one channel and not in the other, or if a speech signal in one channel has an amplitude which is larger by at least a preset threshold than the speech signal in the other channel, the attenuators are so adjusted that the value of the attenuation in said one channel is decreased in a step-wise manner while the value of the attenuation in the other channel is increased in a step-wise manner,

(c) the above threshold being altered in accordance with the level of coupling between the two channels, and

(d) the overall sum of the attenuations in said two channels remains substantially constant.

3. An electronic circuit as claimed in claim 1 or 2, wherein the comparator means is provided by a digital computer such as a microprocessor.

4. An electronic circuit as claimed in claim 1, 2 or 3, wherein when the active or passive conditions of the two channels change, the alternations to the attenuators' settings to increase the attenuation in one of said channels is effected only after the elapsement of a preset time interval.

5. An electronic circuit as claimed in claim 1, 2, 3 or 4, wherein the threshold by which the signal in the transmit channel has to exceed the signal in the receive channel is variable in accordance with the transmit noise level.

6. An electronic circuit as claimed in claim 5, wherein the threshold is dependent on the value of a signal derived from the transmit channel via a low-pass filter.

7. An electronic circuit as claimed in claim 6, wherein said low-pass filter is implemented in a software manner by the microprocessor, and wherein the implementation involves determinations on the bases of a number, e.g. 16, of successive signal samples.

8. An electronic circuit as claimed in any one of claims 1 - 7, and wherein the threshold before a speech signal is detected in the receive channel is automatically adjusted in accordance with receive channel signal level.

9. An electronic circuit as claimed in any one of claims 1 to 8, and wherein the circuitry operations for controlling which channel is active also performs the same functions when the circuit is operating in a non-handsfree mode.

10. An electronic circuit for use in a loudspeaking or handsfree telephone, substantially as described with reference to the accompanying drawings.

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